two channel, matrix encoded audio, the reference does not disclose the particular features recited in the claims of the present application. For example, Claim 1 contains the steps of "subband filtering the two-channel matrix encoded audio into a plurality of two-channel subband audio signals;" and also "synthesizing the mutichannel subband audio signals in the subbands to reconstruct the multichannel audio". Neither reference not discloses subband filtering or synthesizing the subband audio signals; they disclose merely steering audio, with all spectral bands steered in concert.

Claim 1 also includes the step of "<u>separately</u> steering the two channel subband audio signals into a plurality of two-channel <u>subband</u> audio signals." (emphasis added). This step is not taught by the reference. Since the reference does not teach separating each channel into a plurality of subband audio signals, it could not possibly teach "<u>separately</u>" steering the subband audio signals. The Dressler reference discloses diverting bass in a bass management scheme, which does not steer the bass in a sound field, but merely statically assigns it to a bass speaker. Furthermore, Dressler in the encode and decode processes does filter to discriminate a midrange (the 100hz to 7khz region of the spectrum) for steering. The remaining portion of the spectrum is not steered at all. This is easily distinguished from the limitations recited in claim one, which require subband filtering <u>into a plurality of two-channel subband signals</u>, <u>separately steering</u> the two channel subband signals, and <u>synthesizing</u> the subband signals to reconstruct multichannel audio. Only one frequency range is steered in the reference; the others are not steered (thus no "separately steering" step as recited in claim 1). The wide spectrum between 100Hz and 7kHz is not in Dressler subdivided into subbands for separate subband steering. No synthesizing of subband signals into a reconstructed signal takes place in Dressler.

Note that the methods of subband coding and decoding are known as they apply to audio compression for transmission through noisy or band-limited channels. See, for example, Craig Marven, <u>A Simple Approach to Digital Signal Processing</u>, (John Wiley 1996), pages 162-164; Stephen J. Solari, <u>Digital Video and Audio Compression</u>, (McGraw Hill 1997), pages 163-185. The terms "subband filtering" and "synthesizing" have meanings which are well understood from the art of subband coding.

Claims 2, 3, 4, 6, 9 and 11 should be found to have novelty for the reasons set forth in connection with Claim 1. Regarding obviousness, the remarks of the Examiner in connection with Claims 5 and 17 may to be pertinent. Accordingly, the rebuttal found below, in connection with Claims 5 and 17, is also pertinent. Applicant submits that for the reasons set forth below in connection with Claims 5 and 17, the present invention is not obvious over the cited references.

Claim 15 contains limitations similar to those of claim 1: i.e., subband filtering, separately steering the subbands, and synthesizing the subband signals to reconstruct. This claim and those depending therefrom should be found novel and inventive for the same reasons set forth for Claim 1.

The office action also rejects claims 2, 3, 4, 6, 9 and 11 for obviousness under 35 USC 103(a). For the following reasons, the applicant disagrees and requests reconsideration of the rejection.

Rejection of these claims under Section 103 is based on the supposed teaching of the Dressler reference (B). Applicant suggests that the Examiner is reading into Dressler features which are not disclosed therein. In short, Dressler does not disclose any method for treating a plurality of simultaneous dominant signals, merely a means for switching between temporally distinct dominant signals, as dominance shifts from moment to moment. Furthermore, the Dressler reference does not provide any suggestion or motivation to modify the prior art to obtain the invention as claimed, nor can it be combined with any other reference to obtain the invention as claimed.

The correct reading of Dressler (B) can be verified directly from the article itself. On the same page cited by the Examiner (page 7), line 25, the article defines "Dominant sound" as "simply that—the sound that is most prominent in the mix at any given instant in time." Further down, in line 41, the article states "while two different sounds may seem to have the same average loudness, it is likely that, on an instantaneous basis, one of them will be dominant over the other and that the dominance will continuously alternate between them." This makes it clear that Dressler's concept of dominance is an instantaneous one: at any instant there can be only one dominant sound.

Dressler's system is limited to one dominant sound at any instant. The reference to enhancement on an "instantaneous basis" is misleading, but does not indicate the

ability to handle multiple, simultaneous dominant audio signals. Rather, he must have intended instantaneous in the sense of "in close succession" rather than "simultaneous." This is made clear by the sentences beginning at line 51 (immediately following the sentence quoted by the Examiner), "in effect, time-division multiplexing its action among several individual sounds occurring in rapid succession. Even though the decoder is essentially providing directional enhancement for sounds at only one position at a time. . ." The Dressler system is capable of quickly shifting between two dominance positions or of providing a slower speed of switching, according to the nature of the musical content; but it does not reconstruct a signal with a plurality of dominant audio signals at any given instant.

Because Dressler does not recognize the existence of multiple, simultaneous dominant signals (in different subbands), the reference does not and cannot provide any motivation to modify the teachings of Fosgate to handle multiple dominant signals (in different subbands). Nor is there any suggestion that such a modification would provide a more realistic sound (as presumed by the Office action). Dressler's definition of dominance excludes the very possibility of multiple dominant signals at any given instant. Furthermore, the reference does not indicate that there is any shortcoming in the prior method of detecting a single dominant signal at any instant. There is no suggestion to seek multiple dominant signals in different subbands, or any hint that Dressler's method is unsatisfactory.

Similar argument applies to claim 3. Dressler clearly does not disclose a method of decoding wherein the dominant audio signals reside in different subbands, nor would it have been obvious to modify Fosgate to do so. Dressler does not recognize the necessity or desirability of separately steering in each subband, because he does not recognize the existence of a plurality of dominant audio signals, in different subbands. His own definition forbids such recognition. Furthermore, consideration of Dressler's block diagrams confirms that his system cannot handle a plurality of simultaneous dominant signals in different subbands.

Regarding claim 4, the examiner has failed to consider the language "for each subband" (as well as the other limitations from claim 1, discussed previously). No combination of Dressler and Fosgate can produce the present invention, because they

cannot accomplish separate steering in separate subbands. The same argument applies to claims 6-9.

Regarding claim 11, the elements of claim 1 are simply not found in the Fosgate reference, as discussed above.

The office action asserts that Claim 5 is obvious over Davis et al. Applicant disagrees for the following reasons, and requests reconsideration.

The office action asserts that it would have been obvious to employ the bark bands of Davis et al. for the purpose of taking into account the psychoacoustic properties of the audio signal. On the contrary, it is not obvious for several reasons, beginning with the fact that psychoacoustic masking is not relevant to the goals of the present invention. The Examiner correctly points out that subband coding can reduce the amount of information transmitted in a particular frequency band where the resulting coding noise is psychoacoustically masked by neighboring spectral components. Such phenomenon is relevant in an encoder because it facilitates efficient coding for transmission of information through a band limited channel. In contrast, the present invention concerns a decoder or receiver, not an encoder. There is no incentive to utilize psychoacoustic masking or indeed any compression techniques, because the information has already been coded and transmitted before it even reaches the decoder which is the subject of the present claimed invention. It could not have been obvious to employ the bark bands of the Davis patent "for the purpose of taking into account the psychoacoustic properties of the audio signal" (as asserted in the written opinion) because the particular psychoacoustic properties of the audio signal (masking effects) are irrelevant to the decoder of the invention.

Some background on subband coding may help understand the very different purposes of Davis compared to those of the present invention. The typical application of subband coding (and decoding) is for a purpose different from that in the present invention. Usually, subband coding and synthesis is used to reduce the bandwidth required to transmit or store information that has redundancy unevenly distributed through the spectrum. It is not conventionally used to enhance or manipulate the spatial perception of audio, as in the present application. More specifically, consider a typical

application of subband audio coding: Before transmission through a communication (or recording) channel, the audio is subband filtered and the data is decimated. The information is then coded with redundancy reducing techniques (compressed). The compressed data is then transmitted. Upon reception, the data is decoded in each subband. Each subband then goes through a subband synthesizing filter to recover the samples lost through decimation. The subbands are finally mixed to reconstruct the audio (approximately).

By contrast, in the present invention, the filtering occurs in the receiver, after receiving the encoded audio. No compression is necessary. The subbands are separately steered only to enhance spatial perception.

The language quoted from the Davis patent concerns the use of subband coding before transmission, to reduce the amount of information transmitted in a particular frequency band where the resulting coding noise is psychoacoustically masked by neighboring spectral components. As noted above, the present invention does not pertain to the reduction of information to be transmitted; rather, the decoding method of the invention is applied after reception and initial decoding of the audio data in a receiver. Davis does not subband filter the data after receipt and decoding, as presently claimed (claim 1). Rather, the Davis patent assumes that the data has been subband coded (or transform coded) before transmission.

While it may be obvious to break audio into subbands before encoding (to obtain benefits of better compression), it certainly is not obvious to break audio into subbands after transmission, in the decoder, as in the present invention. The claimed invention does not even presuppose any subband coding or subband filtering before transmission. After transmission, the psychoacoustic masking effects, and the advantages of compression, are irrelevant because the critical transmission of data has already been accomplished. In fact, subband filtering at the receiver (after decoding) is inconvenient, unnecessary and unconventional. (Subband filtering should not to be confused with reconstructive subband filtering, or synthesis filtering in a subband decoder, which is a quite different process. )

In short, that which is obvious in a transmitter is not necessarily obvious in a receiver, because their purposes are quite different. It is obvious to affix postage to a

letter before mailing; it is unusual to do so after receipt. Thus, there was no motivation in the prior art to apply the teachings of Davis, and Davis could not be combined with the teachings of Fosgate to produce the claimed invention.

Claims 7 and 8 should be found novel and to involve an inventive step for the same reasons discussed above in connection with claims 5 and 1. Note that claims 7 and 8 also refer to steering of the subband audio signals, not merely steering a composite, wideband signal.

Regarding claim 9, applicant points out that neither reference nor the combination teaches computing a dominance vector for each subband, as claimed. The arguments set forth above in connection with claims 7, 5, and 1 are also applicable to claim 9.

Claims 12-14 have been cancelled.

Claim 15 is discussed above, and should be allowable for the same reasons discussed in connection with Claim 1.

Regarding Claim 16, the examiner has conceded that "Dressler A fails to disclose the method of claim 15 wherein the reconstructed multichannel audio comprises a plurality of dominant signals that reside in different subbands." As discussed above, Dressler B does not admit of the possibility of a plurality of dominant signals at any given instant, thus provides no motivation to modify Dressler A or any other reference to obtain the present invention as claimed. The references simply do not recognize, as the applicant has recognized, that multiple dominant signals may exist at a given time, each being dominant in a given subband and each having distinct spatial location. Thus the references provide no motivation to modify the prior art as in the claimed invention. Furthermore, no combination of Dressler A and Dressler B can produce the claimed invention.

Regarding Claim 17, the remarks above in connection with Claim 5 are pertinent. The teaching of Davis, to reduce the amount of information transmitted in a band when the noise is psychoacoustically masked, is completely irrelevant in a decoder. One does not gain any advantage by compressing the audio after transmission/ reception. It is entirely analogous to locking the barn door after the cows have left the barn. Thus, it would not have been obvious to subband filter the subband signals into a plurality of bark bands. Nor would it have been obvious to "group" the subband signals into such

bark bands, as claimed. Taking into account the psychoacoustic properties of the audio signal does not serve any obvious purpose after receiving and decoding.

Claim 10 was objected to as dependent upon a rejected base claim, but was found otherwise allowable. The claim has not been rewritten in independent form at this time because applicant believes that the Examiner may find applicant's arguments persuasive with respect to the independent claims from which claim 10 depends. In light of recent case law, applicant is compelled to avoid or defer amendments when possible (even purely formal amendments).

Applicant believes that the arguments set forth above provide adequate reason for reconsideration of the rejection of claims 1-19, and that the amended claims should now be allowable. Should any further issues remain unresolved, the Examiner is urged to contact the undersigned attorney by phone to discuss the issues. The applicant believes that such an interview would be most productive in clarifying the issues.

Respectfully submitted,

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